(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) Publication number: **0 588 181 A1**

(12) **EUROPEAN PATENT APPLICATION**

(21) Application number: 93114130.3

(22) Date of filing: 03.09.93

(51) Int. Cl.5: G06F 15/68, H03H 17/02, H04N 5/21

(30) Priority: 14.09.92 EP 92402502

(43) Date of publication of application: 23.03.94 Bulletin 94/12

(84) Designated Contracting States: DE ES FR GB IT

(71) Applicant: THOMSON CONSUMER ELECTRONICS S.A.
9, Place des Vosges,
La Défense 5
F-92400 Courbevoie(FR)

(72) Inventor: Heimburger, Catherine
20D, rue Himmerich
F-67000 Strasbourg(FR)
Inventor: Knee, Michael
131a, route de Lyon
F-67400 Illkirch(FR)

(74) Representative: Einsel, Robert, Dipl.-Ing.
Deutsche Thomson-Brandt GmbH
Patent- und Lizenzabteilung
Göttinger Chaussee 76
D-30453 Hannover (DE)

(54) **Method and apparatus for noise reduction.**

(57) Different kinds of video noise filters are known. Normally such filters need a certain amount of memory, e.g. field memories, or are not suited for every noise level or are not adaptive to the local structure of the picture.
The inventive solution is globally and locally adaptive. A noisy input signal x is filtered with a restoration filter of median type to generate a filtered input signal y. The sum of the absolute differences between filtered and unfiltered signal is calculated for each position of a sliding window within the input signal, representing a local estimate of the noise, and is combined with a global measure of the input signal noise to compute two coefficients a and b which are respectively applied to the unfiltered and filtered signal to generate the output signal z = a*x + b*y.
Advantageously different kind of filters operate in parallel, whereby the kind of filter elected is locally adapted to the picture activity.

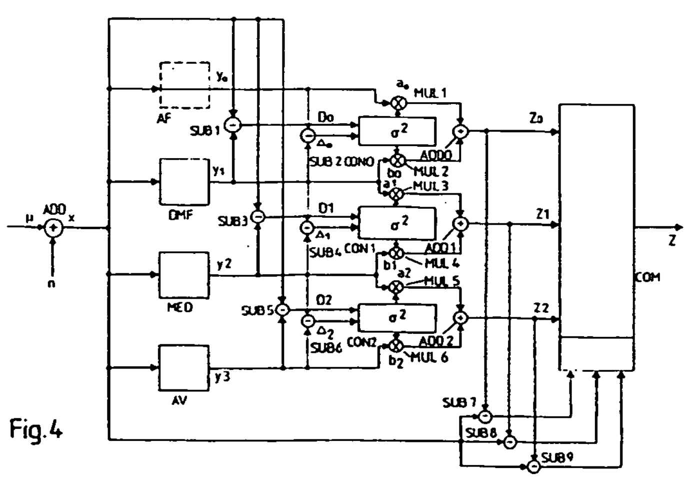



Fig.4

EP 0 588 181 A1

The present invention relates to a method and to an apparatus for noise reduction.

## Background

Different kinds of video noise filters are known. M. Unser, "Improved restoration of noisy images by adaptive least-squares post-filtering", Signal Processing 20 (1990), pp. 3-14, gives an example. Normally such filters need a certain amount of memory, e.g. field memories, or are not suited for every noise level or are not adaptive to the local structure of the picture.

## Invention

It is one object of the invention to disclose a method for noise reduction, which does need only simple hardware and takes into account different noise levels and picture structures. This object is reached by the method disclosed in claim 1.

It is a further object of the invention to disclose an apparatus which utilises the inventive method. This object is reached by the apparatus disclosed in claim 9.

For the inventive noise reduction the main requirements listed below are considered:

* field processing i.e. without field memory;
* few memories and simple hardware realisation;
* processing adaptive to any level of noise;
* processing adaptive to the local structure of the picture,

i.e. detecting plain areas, edges and fine details so as to preserve the relevant information of the picture.

Different filters could be used:

1. linear low pass filter with different coefficients;
2. median filter;
3. linear filter in the direction of the edges (directional filter);
4. directional median filter;
5. weighted averages of filters 1, 2, 3 and 4;
6. filtering with one of the filters 1, 2, 3 and 4 with fixed threshold and local measure of picture activity by the Laplacian.

These filter types have the following properties:

* median filters are more suitable to remove noise in plain areas than linear filters especially in the case of high frequency noise of high amplitude;
* median filters preserve edges and linear filters do not;
* directional filters are very efficient for preserving edges and picture accuracy while filtering slightly but efficiently enough;
* median filters completely lose the fine textures (of a zone plate test picture, for example) and the only way to preserve them is to use a directional filter;
* weighted averages of filters give good results but the weighting coefficients are not easy to adapt locally without correct noise estimation and can lead to hard switching and to the introduction of artefacts;
* thresholds and local measures of activity give good results when adapted to the picture but they have to be combined with noise estimation on the whole picture.

The inventive solution is globally and locally adaptive. The implemented filter is derived from that proposed by Unser (cited above) and works as follows:

The noisy input signal x is filtered with a restoration filter of median type to generate a filtered input signal y. The sum of the absolute differences between filtered and unfiltered signal is calculated for each position of a sliding window within the input signal, representing a local estimate of the noise, and is combined with a global measure of the input signal noise to compute by least squares regression two coefficients a and b which are respectively applied to the unfiltered and filtered signal to generate the output signal $z = a^*x + b^*y$.

Advantageously the only required prior information is the noise variance $\sigma^2$, assuming that the original signal $\mu$ is degraded by additive stationary noise n. The size of the local noise estimation window can be different from the size of the window used by the restoration filter, wherefore they can be adjusted independently for optimal performance.

The weighting coefficients a and b are made optimal in the sense that they are minimising the quadratic error $\epsilon^2$ between the output signal z and the noise free original signal $\mu$. It is true that the original signal is not known at a receiver, but global noise statistics can be calculated at the receiver side (e.g. described in

EP-A-92400785) and used for that optimisation. Preferably a + b = 1, so that no bias is introduced by the filtering. This solution usually performs slightly worse, but is rather simple to implement, no computational complexity need be added.
If the filtering tends to degrade the signal, a predominant weight will be given to the unfiltered signal x. Conversely, when the sum of differences is small and close to a reference value corresponding to the residue of the noise alone, the weight will be shifted to the filtered signal.

The choice of the kind of restoration filter does heavily influence the merit of the inventive noise reduction. Therefore, advantageously different kind of filters can operate in parallel, whereby the kind of filter elected has to be locally adapted to the picture activity. That branch which results in a minimum error between the filtered and the unfiltered signal will be selected.

In principle the inventive method is suited for noise reduction, in which for estimating a local noise value within a sliding window of a picture the quadratic error between an unfiltered noisy input signal and the filtered input signal is computed and used for calculating a weighted average of the filtered and unfiltered input signal as output signal, whereby for calculating said weighted average also a global noise value is used and for computing said local noise value pixel difference signals of said window are calculated.

Advantageous additional embodiments of the inventive method are resulting from the respective dependent claims.

In principle the inventive apparatus includes:

- a first filter, a second filter and a third filter having a common input signal and a different kind of characteristic;
- a first control circuit controlling the weighting values of a first multiplier operating on said input signal and a second multiplier operating on the output signal of said first filter, whereby the outputs of these multipliers are combined in first adding means;
- a second control circuit controlling the weighting values of a third multiplier operating on the output signal of said first filter and a fourth multiplier operating on the output signal of said second filter, whereby the outputs of these multipliers are combined in second adding means;
- a third control circuit controlling the weighting values of a fifth multiplier operating on the output signal of said second filter and a sixth multiplier operating on the output signal of said third filter, whereby the outputs of these multipliers are combined in third adding means;
- first subtraction means which feed the difference signal between said input signal and the output signal of said first filter to said first control circuit;
- second subtraction means which feed the difference signal between said input signal and the output signal of said second filter to said second control circuit;
- third subtraction means which feed the difference signal between the output signal of said first filter and the output signal of said second filter to said second control circuit;
- fourth subtraction means which feed the difference signal between said input signal and the output signal of said third filter to said third control circuit;
- fifth subtraction means which feed the difference signal between the output signal of said second filter and the output signal of said third filter to said third control circuit;
- further subtraction means which feed the respective difference signals between said input signal and the output signals of said adding means to a control and mixing circuit which selects an output signal of said adding means according to the minimum error derived from these difference signals and according to a global noise variance value,

whereby for each of the branches said weighting values are derived from error signals calculated in the respective control circuit from said difference signals and from said global noise variance value.

Advantageous additional embodiments of the inventive apparatus are resulting from the respective dependent claim.

## Drawings

Preferred embodiments of the invention are described with reference to the accompanying drawings, which show in:

Fig. 1 first embodiment of an inventive noise reduction circuit;
Fig. 2 filter-related directions in the picture;
Fig. 3 window for noise filtering;
Fig. 4 second embodiment of an inventive noise reduction circuit;
Fig. 5 correction terms for the comparison of the filters errors.

Preferred embodiments

In Fig. 1 noise n is added in a channel, e.g. a TV channel, to a noise free original signal $\mu$, e.g. a TV signal. The resulting noisy input signal x of the inventive apparatus, e.g. in a TV receiver or a VCR, is filtered with a restoration filter MF of median type to generate a filtered input signal y. The difference $\Delta$ between signal x and signal y is calculated in a subtractor SUB and fed to a control circuit CON which forms from $\Delta$ the sum of the absolute differences between filtered and unfiltered input signal for each position of a sliding window within the input signal x, representing a local estimate of the noise.
The global noise statistics can be calculated in a noise measurement circuit NM. The local noise estimate is combined in control circuit CON with the global measure $\sigma^2$ of the input signal noise to compute by least squares regression two coefficients a and b which are respectively applied to the unfiltered and filtered signal to generate the output signal $z = a^*x + b^*y$. This can be done by multiplying signal x in a first multiplier MUL1 by factor a and by multiplying signal y in a second multiplier MUL2 by factor b. The outputs of MUL1 and MUL2 are added in adder ADD0, resulting in a noise-reduced output signal z.

Because the kind of filter MF influences the resulting noise reduction and picture quality, advantageously different filter types can operate in parallel. Then the problem is, how to select the output of the best-fitting filter. In Fig. 4 three filters DMF, MED and AV of different kind are operating, whereby weighted averages of each two consecutive filter outputs are calculated.
Each average is obtained as in Fig. 1 by minimising the squared error $\epsilon^2$ between the output signal z and the noise-free signal $\mu$. A respective filtered output signal $z_i$, i=0 to 2, is selected by examining the respective errors: the selected branch corresponds then to the minimum error. This method allows switching between filters depending on the statistical properties of the local filtered and unfiltered pictures.
The three restoration filters are:

- DMF - directional median filter;
- MED - median filter;
- AV - average.

The sliding window 30 depicted in Fig. 3 has a size of 5 pixels by 3 lines. As the processing is not on a frame basis but on a field (F1; F2) basis, this size corresponds to a region of 5 by 5 pixels for the interlaced picture (lines of field F1).
The directional median filter DMF works as follows:
Let $x_{ji}$, j=0 to 2, i=0 to 4, be the pixels taken into account by window 30. The tested filter directions in Fig. 2 are called $d_i$, i=0 to 5. The related gradients are respectively estimated by:

$$g_0 = |x_{00}-x_{12}| + |x_{24}-x_{12}|;$$
$$g_1 = |x_{01}-x_{12}| + |x_{23}-x_{12}|;$$
$$g_2 = |x_{02}-x_{12}| + |x_{22}-x_{12}|;$$
$$g_3 = |x_{03}-x_{12}| + |x_{21}-x_{12}|;$$
$$g_4 = |x_{04}-x_{12}| + |x_{20}-x_{12}|;$$
$$g_5 = 1/2^*(|x_{10}-x_{12}| + |x_{11}-x_{12}| + |x_{13}-x_{12}| + |x_{14}-x_{12}|).$$

The selected direction is given by the smallest gradient. The respective output signals $y_1$ of filter DMF are median values:

$$m_0 = \text{median}(x_{00}, x_{12}, x_{24});$$
$$m_1 = \text{median}(x_{01}, x_{12}, x_{23});$$
$$m_2 = \text{median}(x_{02}, x_{12}, x_{22});$$
$$m_3 = \text{median}(x_{03}, x_{12}, x_{21});$$
$$m_4 = \text{median}(x_{04}, x_{12}, x_{20});$$
$$m_5 = \text{median}(x_{11}, x_{12}, x_{13}).$$

If two or more of these directions result in a minimum gradient, the average of the respective signals is taken to generate the output $y_1$. Such multiple minimum gradients appear at less than 1% of pixels.
The output $y_2$ of filter MED is:

$$y_2 = \underset{\substack{0 \le i \le 4 \\ 0 \le j \le 2}}{\text{median}} (x_{ji})$$

The output $y_3$ of filter AV is:

$$y_3 = \frac{1}{15} * \sum_{\substack{0 \le i \le 4 \\ 0 \le j \le 2}} (x_{ji})$$

In Fig. 4 the circuits ADD, DMF, SUB2, CON0, MUL1, MUL2 and ADD0 correspond to the circuits ADD, MF, SUB, CON, MUL1, MUL2 and ADD of Fig. 1. The input of filter DMF is also connected to a second filter MED and to a third filter AV and can also be connected to an additional filter AF which forms the output signal $y_0$ from the noisy input signal x. The input signal x is also fed to a first, a third and a fifth subtractor SUB1, SUB3 and SUB5. The output signal $y_1$, $y_2$ and $y_3$, respectively, of filter DMF, MED and AV, respectively, is subtracted in the subtractor SUB1, SUB2 and SUB3, respectively from the input signal x. The outputs $D_0$, $D_1$, and $D_2$, respectively, of these subtractors are each connected to a respective control circuit CON0, CON1 and CON2. The difference signal $\Delta_0$, $\Delta_1$ and $\Delta_2$, respectively, calculated in a second, a fourth and a sixth subtractor SUB2, SUB4 and SUB6, respectively, between signals $y_0$ (=x) and $y_1$, $y_1$ and $y_2$, and $y_2$ and $y_3$, respectively, are also fed to the according control circuit CON0, CON1 and CON2. If filter AF is omitted, SUB1 can also be omitted because $D_0 = \Delta_0$.
The first control circuit CON0 controls the coefficient $a_0$ of a first multiplier MUL1 which multiplies signal $y_0$ and the coefficient $b_0$ of a second multiplier MUL2 which multiplies signal $y_1$. The outputs of these multipliers are added in adder ADD0, forming output signal $z_0$.
The second control circuit CON1 controls the coefficient $a_1$ of a third multiplier MUL3 which multiplies signal $y_1$ and the coefficient $b_1$ of a fourth multiplier MUL4 which multiplies signal $y_2$. The outputs of these multipliers are added in adder ADD1, forming output signal $z_1$.
The third control circuit CON2 controls the coefficient $a_2$ of a fifth multiplier MUL5 which multiplies signal $y_2$ and the coefficient $b_2$ of a sixth multiplier MUL6 which multiplies signal $y_3$. The outputs of these multipliers are added in adder ADD2, forming output signal $z_2$.
In a seventh, eighth and ninth subtractor SUB7, SUB8 and SUB9, respectively, the difference signals between the according output signals $z_0$, $z_1$ and $z_2$ and the input signal x (error on window) are calculated. These subtractor output signals are used in a fourth control and mixing circuit COM for connecting one of the output signals $z_0$, $z_1$ and $z_2$ with output z, according to the minimum error.
For calculating the minimum error in the four control circuits, the known noise variance $\sigma^2$, as described for Fig. 1, is used. Hard or soft switching can be made in control circuit COM.

The quadratic errors to be minimised in CON0, CON1 and CON2 are for i=0 to 2:

$$\begin{aligned} \varepsilon_i &= 1/N_R * \sum_R (a_i y_i + b_i y_{i+1} - \mu)^2 \qquad (1) \\ &= 1/N_R * \sum_R (z_i - \mu)^2 \\ &= 1/N_R * S_{z_i - \mu, z_i - \mu}, \end{aligned}$$

where $N_R$ is the total number of pixels in the region R which corresponds to the sliding window 30. The notation $S_{u,v}$ is used to designate a sum of squares over the region R and is defined as:

$$Su,v = \sum_R u(k,l)v(k,l),$$

where k and l are elements of R and u and v stand either for x, $Y_i$, $z_i$ or $\mu$. $\mu$ is not known so that $Sy_i,\mu$ cannot be computed in such a way. Instead, use is made of an a prior knowledge of the noise statistics. It is assumed for simplicity that $a_i$ and $b_i$ are constant over the region R. Then (equation 1):

$$N_R\,\varepsilon_i = a_i^2\sum_R y_i^2 + b_i^2\sum_R y_{i+1}^2 + \sum_R \mu^2 + 2a_i b_i \sum_R y_i y_{i+1} - 2a_i \sum_R y_i \mu - 2b_i \sum_R y_{i+1}\mu$$

$$= a_i^2 Sy_i,y_i + b_i^2 Sy_{i+1},y_{i+1} + S\mu,\mu + 2a_i b_i Sy_i,y_{i+1} - 2a_i Sy_i,\mu - 2b_i Sy_{i+1},\mu$$

The constraint is $a_i + b_i = 1$. The minimisation with this constraint leads to the following system of equations:

$$a_i Sy_i,y_i + b_i Sy_i,y_{i+1} + \lambda_i = Sy_i,\mu$$
$$a_i Sy_i,y_{i+1} + b_i Sy_{i+1},y_{i+1} + \lambda_i = Sy_{i+1},\mu$$
$$a_i + b_i = 1,$$

where $\lambda_i$ is a Lagrange multiplier.

The solution is:

$a_i = ( Sy_i,\mu - Sy_{i+1},\mu - Sy_i,y_{i+1} + Sy_{i+1},y_{i+1})/\Delta_i$ $a_i = ( Sy_i,\mu - Sy_{i+1},\mu - Sy_i,y_{i+1} + Sy_{i+1},y_{i+1})/\Delta_i$
$b_i = (-Sy_i,\mu + Sy_{i+1},\mu - Sy_i,y_{i+1} + Sy_i,y_i )/\Delta_i$
where $\Delta_i = Sy_i,y_i - 2Sy_i,y_{i+1} + Sy_{i+1},y_{i+1} = Sy_i - y_{i+1},y_i - y_{i+1}$

Because $\mu = x - n$,
$Sy_i,\mu = Sy_i,x - Sy_i,n = Sx,y_i - S\mu_i',n - Sn_i',n = Sx,y_i - \rho_i\sigma^2$ where for each filter $F_i$: $\mu_i'$ is the filtered noise free signal, $n_i'$ is the filtered noise, $\sigma^2 = \mathbf{E}(n^2(k, l))$ is the known noise variance and $\rho_i = \mathbf{E}(n(k, l)n_i'(k, l))$ is the residual noise correlation coefficient after filtering with $F_i$ (i.e. $\rho = 1$: no filtering; $\rho = 0$: full filtering, no noise left).
The assumption of stationary noise allows here easy estimation of these coefficients $\rho_i$. But if they are known for another kind of noise, it is possible to extend the filter to less restrictive conditions.
Using these coefficients, $a_i$ and $b_i$ can be estimated to:

$$a_i = ( Sx,y_i - Sx,y_{i+1} - Sy_i,y_{i+1} + Sy_{i+1},y_{i+1} - \rho_i\sigma^2 + \rho_{i+1}\sigma^2)/\Delta_i$$
$$b_i = (-Sx,y_i + Sx,y_{i+1} - Sy_i,y_{i+1} + Sy_i,y_i + \rho_i\sigma^2 - \rho_{i+1}\sigma^2)/\Delta_i$$

or

$$a_i = (Sx - y_{i+1},y_i - y_{i+1} - \sigma^2(\rho_i - \rho_{i+1}))/Sy_i - y_{i+1},y_i - y_{i+1}$$
$$b_i = (Sy_i - x,y_i - y_{i+1} + \sigma^2(\rho_i - \rho_{i+1}))/Sy_i - y_{i+1},y_i - y_{i+1}$$

Therefore $a_i$ and $b_i$ are known once the correlation term $Sx - y_{i+1},y_i - y_{i+1}$ has been computed within the control circuits CON0, CON1 and CON2 for the current position of the window 30. The output signals $D_0$, $D_1$ and $D_2$, respectively, of the according subtractors SUB1, SUB2 and SUB3 are used for computing these correlation terms. For good stability of the system it is required that $z_i$ is always between $y_i$ and $y_{i+1}$, i.e. $0 \le a_i \le 1$.

For selecting in control and mixing circuit COM the optimum filter branch, it is possible to derive from the values of a and b the quadratic errors:

$$N_R\,\epsilon_i = Sz_i - \mu,z_i - \mu = Sz_i,z_i - 2Sz_i,\mu + S\mu,\mu$$

where

$$S_{\mu,\mu} = S_{x-n,\mu} = S_{x,\mu} - S_{n,\mu} = S_{x,x} - S_{x,n} = S_{x,x} - S_{n,n} - S_{\mu,n} = S_{x,x} - \sigma^2$$

and

$$S_{z_i,\mu} = S_{a_i y_i + b_i y_{i+1},\mu} = a_i S_{y_i,\mu} + b_i S_{y_{i+1},\mu} = a_i S_{x,y_i} - a_i \rho_i \sigma^2 + b_i S_{x,y_{i+1}} - b_i \rho_{i+1} \sigma^2 = S_{x,z_i} - \sigma^2 (a_i \rho_i + b_i \rho_{i+1}).$$

Therefore

$$N_R \epsilon_i = S_{z_i,z_i} - 2S_{x,z_i} + 2\sigma^2 (a_i \rho_i + b_i \rho_{i+1}) + S_{x,x} - \sigma^2 = S_{z_i-x,z_i-x} + \sigma^2 (2a_i \rho_i + 2b_i \rho_{i+1} - 1).$$

The correction term $e_i = \sigma^2(2a_i\rho_i + 2b_i\rho_{i+1}-1)$ can be explained by the fact that the more a given filter smooths the picture, the higher the term $S_{z_i-x,z_i-x}$ will be. Therefore, before the terms generated by filters of different efficiencies can be compared, they must be compensated by a specific kind of function.
The graph of this function is given in Fig. 5 to illustrate this effect when using the current filter type (AV, MED, DMF, AF) parameters. The correction term $e_i$ is shown in relation to coefficient $a_i$.

The filter AV of type 'average' is well suited for plain areas. The two median filters DMF and MED preserve the edges and a good resolution.
To improve the contrast, enhancement processing can be implemented after the noise reduction processing.
If the window 30 is enlarged, also other picture degradations can be reduced, for example blurs and stripes. Advantageously due to field processing no blurs are introduced on moving objects.

In order to reduce the complexity and the costs of the hardware implementation it is possible to modify the described filters in the following way:

* the size of the window for error estimation can be reduced to 5 pixels by 1 line: therefore no additional line memories will be needed (only 2 line memories for the input filters);
* filter DMF can be a directional median filter with only four directions instead of six: $d_0$, $d_2$, $d_4$ and $d_5$;
* filter MED can be a median with a window of three pixels by three lines;
* limitation of the number of bits for the computation of the weighting coefficients (4 bits, for example).

These simplifications have the following influences on the processed pictures:

* if the size of the estimation window is smaller, the simplified filter will tend to filter more than the non simplified one, particularly when there are local horizontal line structures, because no care is taken of the previous and following lines even if they are very different in their local structure;
* simplifications of the input filters do not create too many degradations on the output signal since the filtering is adaptive and always the best solution between the three outputs $z_0$, $z_1$ and $z_2$ is chosen;
* the reduction of the number of bits of the weighting coefficients creates no significant loss of accuracy: 1 or 2 units for an amplitude range of 256.

The big advantage of this new filter is to allow with very few memories and simple hardware implementation a noise filtering that is adaptive simultaneously locally and globally.
Because of this double ability of adaptation the filtering is optimised even with simple filters at the input.

The inventive noise reduction can be used in any video or audio processing devices, e.g. TV receivers and VCR's.

## Claims

1. Method for noise reduction, in which for estimating a local noise value within a sliding window (30) of a picture the quadratic error between an unfiltered noisy input signal (x) and the filtered (MF) input signal (y) is computed and used for calculating (CON) a weighted (a, b) average of the filtered and unfiltered input signal as output signal (z), **characterised in** that for calculating (CON; CON0, CON1, CON2) said weighted (a, b; $a_0$, $b_0$, $a_1$, $b_1$, $a_2$, $b_2$) average also a global noise value (NM, $\sigma^2$) is used, whereby for computing said local noise value pixel difference signals ($\Delta$; $\Delta_0$, $\Delta_1$ $\Delta_2$) of said window are calculated (CON; CON0, CON1, CON2).

2. Method according to claim 1, **characterised in** that said filtering is carried out in two or more branches of different filter characteristics (DMF, MED, AV) each having a weighted average output signal ($z_0$, $z_1$, $z_2$) derived from the filtered input signals ($y_0$, $y_1$; $y_1$, $y_2$; $y_2$, $y_3$) of each two consecutive branches, whereby for computing said local noise value in each of said branches additionally respective

pixel difference signals between each two consecutive of said filtered input signals ($y_1$, $y_2$; $y_2$, $y_3$) are used and for obtaining the final output signal (z) that branch output signal ($z_0$, $z_1$, $z_2$) is selected (COM) which has a minimum error compared to said input signal (x).

3. Method according to claim 2, **characterised in** that when comparing said minimum error, a correction term related to the respective filter characteristic is added to each of the branch errors.

4. Method according to any of claims 1 to 3, **characterised in** that the sum of the two weighting factors ($a_0$, $b_0$; $a_1$, $b_1$; $a_2$, $b_2$) for said weighted average/s is 1.

5. Method according to claim 1 and 4, **characterised in** that the one weighting factor (a) related to said unfiltered input signal (x) of the two (a, b) weighting factors for said weighted average is in the range 0 to 1.

6. Method according to claims 4 and 5 and to claims 2 or 3, **characterised in** that the respective one ($a_0$, $a_1$, $a_2$) of each two ($a_0$, $b_0$; $a_1$, $b_1$; $a_2$, $b_2$) weighting factors for said weighted average is in the range 0 to 1.

7. Method according to any of claims 2 to 4 and 6, **characterised in** that at minimum two of the following filter characteristics are used in said branches: directional median filter (DMF), median filter (MED), average (AV), whereby for said directional median filter at minimum four directions ($d_0$-$d_5$) are evaluated in said sliding window (30).

8. Method according to any of claims 1 to 7, **characterised in** that said sliding window (30) has a size of five pixels by one to five lines when related to a frame basis.

9. Apparatus for a method according to any of claims 2 to 4 and 6 to 8, including:
   - a first filter (DMF), a second filter (MED) and a third filter (AV) having a common input signal (x) and a different kind of characteristic;
   - a first control circuit (CON0) controlling the weighting values ($a_0$, $b_0$) of a first multiplier (MUL1) operating on said input signal (x) and a second multiplier (MUL2) operating on the output signal ($y_1$) of said first filter, whereby the outputs of these multipliers are combined in first adding means (ADD0);
   - a second control circuit (CON1) controlling the weighting values ($a_1$, $b_1$) of a third multiplier (MUL3) operating on the output signal ($y_1$) of said first filter and a fourth multiplier (MUL4) operating on the output signal ($y_2$) of said second filter, whereby the outputs of these multipliers are combined in second adding means (ADD1);
   - a third control circuit (CON2) controlling the weighting values ($a_2$, $b_2$) of a fifth multiplier (MUL5) operating on the output signal ($y_2$) of said second filter and a sixth multiplier (MUL6) operating on the output signal ($y_3$) of said third filter, whereby the outputs of these multipliers are combined in third adding means (ADD2);
   - first subtraction means (SUB2) which feed the difference signal ($\Delta_0$) between said input signal (x) and the output signal ($y_1$) of said first filter to said first control circuit (CON0);
   - second subtraction means (SUB3) which feed the difference signal ($D_1$) between said input signal (x) and the output signal ($y_2$) of said second filter to said second control circuit (CON1);
   - third subtraction means (SUB4) which feed the difference signal ($\Delta_1$) between the output signal ($y_1$) of said first filter and the output signal ($y_2$) of said second filter to said second control circuit (CON1);
   - fourth subtraction means (SUB5) which feed the difference signal ($D_2$) between said input signal (x) and the output signal ($y_3$) of said third filter to said third control circuit (CON2);
   - fifth subtraction means (SUB6) which feed the difference signal ($\Delta_2$) between the output signal ($y_2$) of said second filter and the output signal ($y_3$) of said third filter to said third control circuit (CON2);
   - further subtraction means (SUB7, SUB8, SUB9) which feed the respective difference signals between said input signal (x) and the output signals of said adding means (ADD0, ADD1, ADD2) to a control and mixing circuit (COM) which selects an output signal ($z_0$, $z_1$, $z_2$) of said adding means according to the minimum error derived from these difference signals and according to a global noise variance value ($\sigma^2$).

whereby for each of the branches said weighting values are derived from error signals calculated in the respective control circuit from said difference signals and from said global noise variance value ($\sigma^2$).

10. Apparatus according to claim 9, **characterised in** that said first filter (DMF) is a directional median filter wherein at minimum four directions ($d_0$-$d_5$) are evaluated and said second filter (MED) is a median filter and said third filter (AV) is an averaging filter.

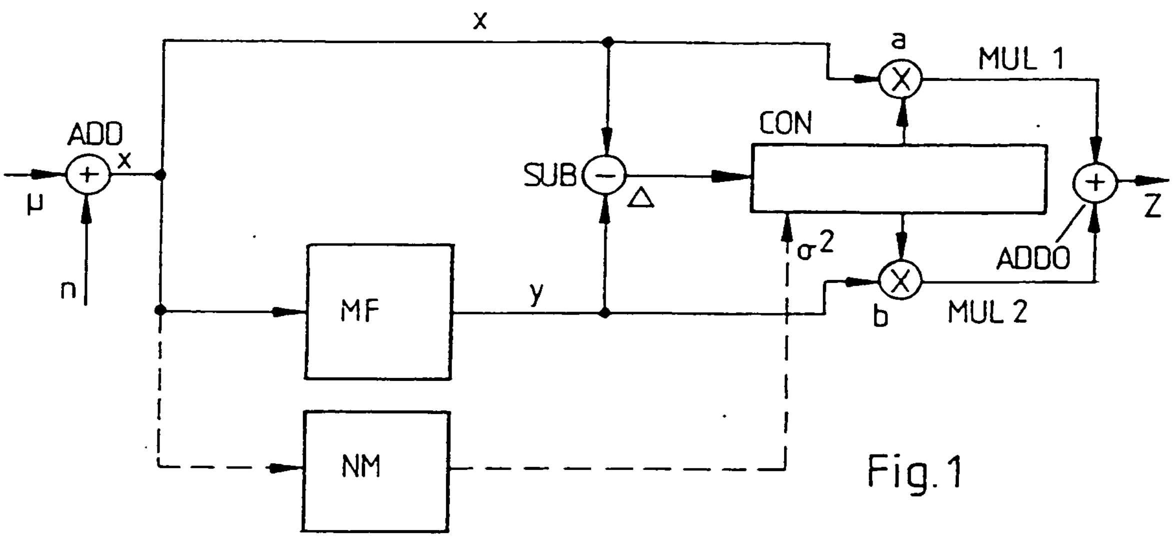



Fig.1

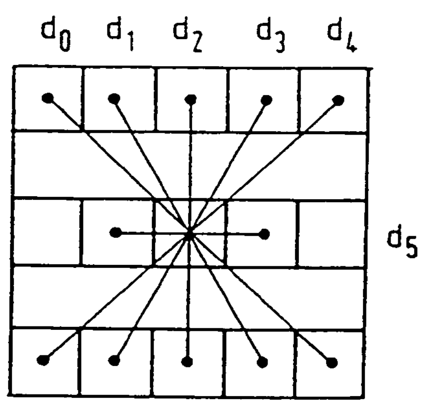



Fig.2

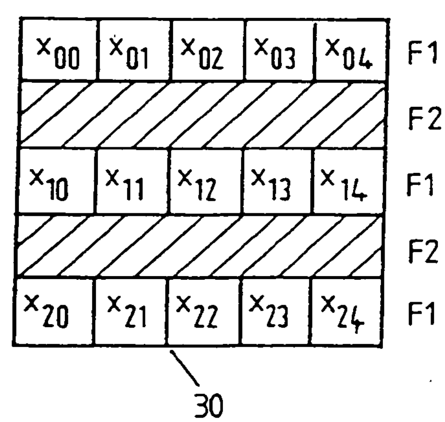



Fig.3

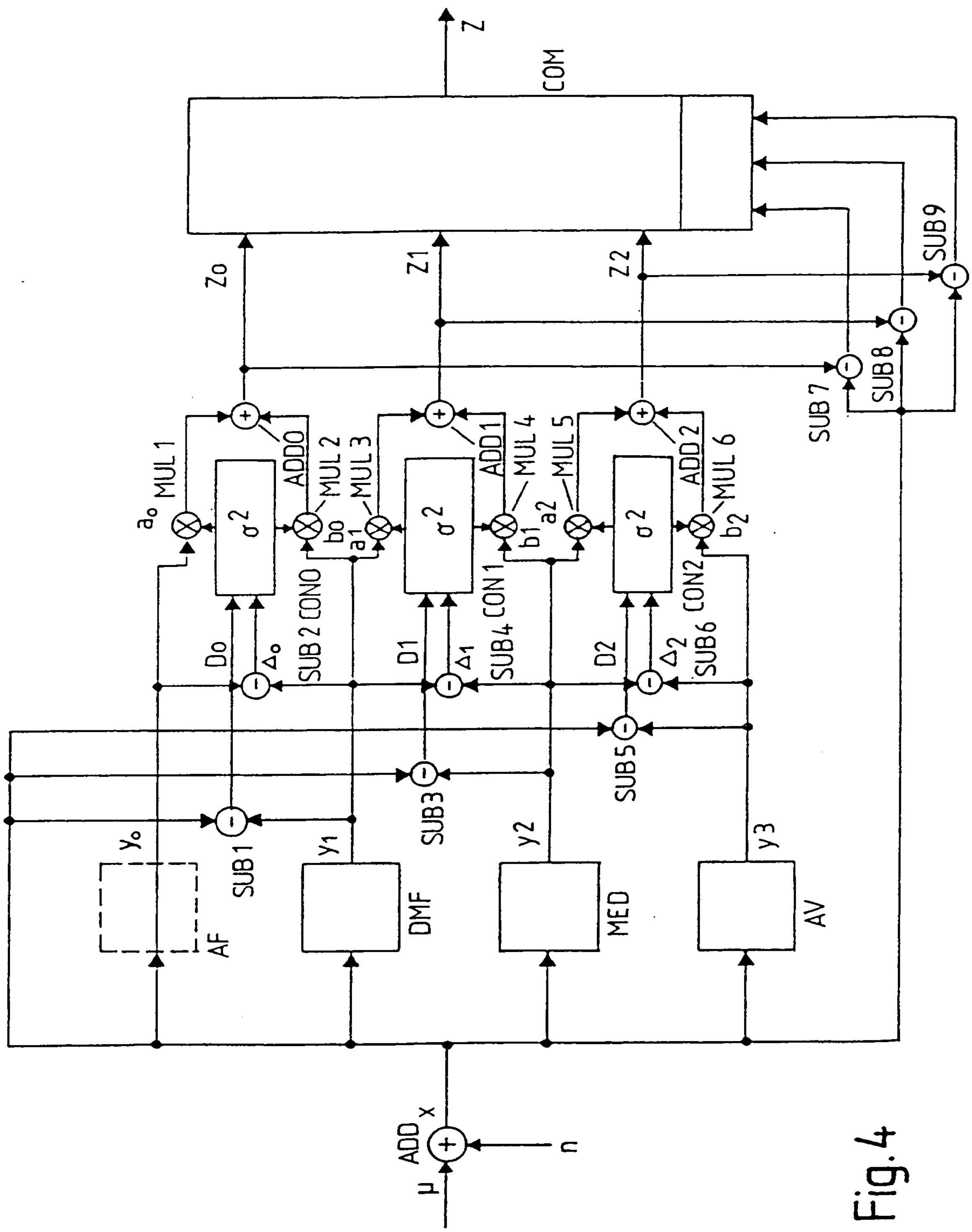
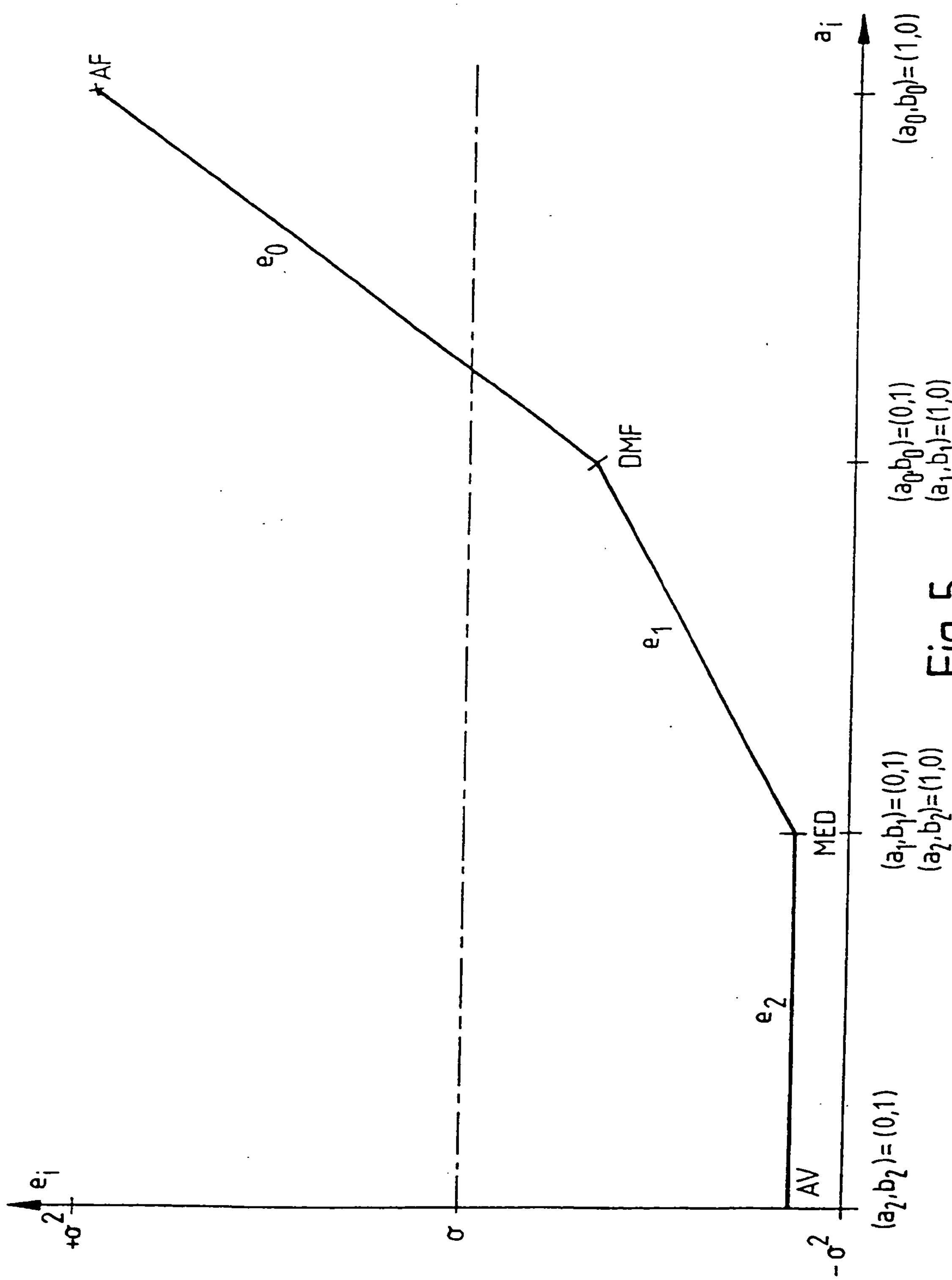

Z
COM
Zo
Z1
Z2
SUB7
SUB8
SUB9
$a_o$ MUL 1
ADD0
MUL 2
MUL 3
ADD1
MUL 4
MUL 5
ADD 2
MUL 6
$\sigma^2$
$b_o$
a1
$b_1$
a2
$b_2$
SUB 2 CON0
SUB4 CON1
SUB6 CON2
Do
D1
D2
$\Delta_o$
$\Delta_1$
$\Delta_2$
SUB1
SUB3
SUB5
$y_o$
y1
y2
y3
AF
DMF
MED
AV
ADD
μ
x
n

Fig.4



$e_i$
$a_i$
$+\sigma^2$
$\sigma$
$-\sigma^2$
AF
DMF
MED
AV
$e_0$
$e_1$
$e_2$
$(a_0,b_0)=(1,0)$
$(a_0,b_0)=(0,1)$
$(a_1,b_1)=(1,0)$
$(a_1,b_1)=(0,1)$
$(a_2,b_2)=(1,0)$
$(a_2,b_2)=(0,1)$


Fig. 5

European Patent Office

# EUROPEAN SEARCH REPORT

Application Number

EP 93 11 4130

## DOCUMENTS CONSIDERED TO BE RELEVANT

| Category | Citation of document with indication, where appropriate, of relevant passages | Relevant to claim | CLASSIFICATION OF THE APPLICATION (Int. Cl.5) |
|---|---|---|---|
| Y<br>A | US-A-4 334 244 (CHAN ET AL.)<br>* column 2, line 51 - column 4, line 10; figure 2 *<br>--- | 1,4,5<br>9 | G06F15/68<br>H03H17/02<br>H04N5/21 |
| Y,D<br><br><br><br><br><br><br>A | SIGNAL PROCESSING. EUROPEAN JOURNAL DEVOTED TO THE METHODS AND APPLICATIONS OF SIGNAL PROCESSING<br>vol. 20, no. 1, May 1990, AMSTERDAM NL<br>pages 3 - 14<br>UNSER 'Improved restoration of noisy images by adaptive least-squares post-filtering'<br>* page 4, column 2, line 39 - page 7, column 1, line 24; figure 2 *<br>--- | 1,4,5<br><br><br><br><br><br><br>9 | |
| A | IEEE TRANSACTIONS ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING<br>vol. 37, no. 8, August 1989, NEW YORK US<br>pages 1293 - 1298<br>KUNDU ET AL. 'Double.window Hodges-Lehman (D) filter and hybrid D-median filter for robust image smoothing'<br>* page 1294, column 1 - column 2, line 36; table 1 *<br>----- | 2,7,8 | TECHNICAL FIELDS SEARCHED (Int. Cl.5)<br>H04N<br>H03H<br>G06F |

The present search report has been drawn up for all claims

| Place of search | Date of completion of the search | Examiner |
|---|---|---|
| BERLIN | 15 DECEMBER 1993 | MATERNE A. |

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P0401)